# VoIP versus VoMPLS Performance Evaluation

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**Abstract**

Increasing number of Internet users forced them to use popular services, television and telephone, to use the Internet as a medium to reach their customers. Organizations are now applying VoIP and MPLS for providing the convergence of voice and data communications over single network infrastructure. VoIP (Voice over Internet Protocol) usage all over their branches, this reduces a huge amount of cost for their infrastructure; easily exchanging for voice, video and Data. MPLS (Multiprotocol Label Switching) concept is to standardize a number of multilevel switching solutions. MPLS is the latest step in the evolution of technology switching / routing Internet using a control solution that integrates both IP routing and switching as data link level (level 2) of the OSI (Open Systems Interconnection) model. The performance of the two network infrastructure models will performed and compared one for IP and the other for MPLS, the results came encouraging for the MPLS model.

***Keywords:*** *MPLS; VOIP; QoS; ( Switching / Routing); OPNET Modeler 14.5.*

## 1. Introduction

### 1.1 VoIP Overview

The increase in capacity of the Internet in addition to popularity gives an increasing need to provide real-time voice and video services to the network. VoIP uses the Internet Protocol (IP) to transmit voice as packets over an IP network. By this technology the voice signal firstly digitized, compressed and converted to IP packets and after that it will be transmitted over the IP network. The potential for very low-cost or free voice can be achieved [1].

### 1.2 VoIP control Signaling Protocols

The various signaling protocols, allowing users of VoIP to connect their phone calls. There are several different types of VoIP signaling available today, including H.323, SIP, MGCP,SCCP, MEGACO, and SIGTRAN; but the most prevalent types of signaling protocols today are H.323 and SIP.

- H.323. This is the ITU-T's (International Telecommunications Union) standard that vendors should comply while providing Voice over IP service. This recommendation provides the technical requirements for voice communication over LANs. It was originally developed for multimedia conferencing on LANs, but was later extended to cover Voice over IP. The first version was released in 1996 while the second version of H.323 came into effect in January 1998. The standard encompasses both point to point communications and multipoint conferences [2].
- Session Initiation Protocol (SIP). It is an application layer control protocol for creating, modifying and terminating sessions with one or more participants. The architecture of SIP is similar to that of HTTP (client-server protocol). Requests are generated by the client and sent to the server. The server processes the requests and then sends a response to the client. A request and the responses for that request make a transaction. SIP has INVITE and ACK messages which define the process of opening a reliable channel over which call control messages may be passed. SIP makes minimal assumptions about the underlying transport protocol. This protocol itself provides reliability and does not depend on Transmission Control Protocol (TCP) for reliability. SIP depends on the Session Description Protocol (SDP) for carrying out the negotiation for codec identification. SIP supports session descriptions that allow participants to agree on a set of compatible media types. It also supports user mobility by redirecting requests to the user's current location. The services that SIP provide include :
  - Call Setup: ringing and establishing call parameters at both called and calling party.

- User Availability: determination of the willingness of the called party to engage in communications.
- User Capabilities: determination of the media and media parameters to be used.
- Call handling: the transfer and termination of calls [3].

### 1.3 MPLS Overview

MPLS is a tunneling technology used in many service provider networks. The main functionality is to attach a short fixed-label to the packets that enter into MPLS domain. Label is placed between Layer-2 (Data Link Layer) and Layer-3 (Network Layer) of the packet to form Layer-2.5 label switched network on layer-2 switching functionality without layer 3 IP routing. Packets in the MPLS network are forwarded based on these labels. MPLS provides high performance packet control and forwarding mechanism for routing the packets in the data networks[4].

### 1.4 MPLS Signaling Protocols

IP packets were forwarded looking into its destination address at every router in the path. The packets were forwarded based on the shortest path metric, which is the cost calculated using the time it takes to reach the next hop. When the traffic in the network increases, the link with shortest path becomes heavily congested while the links with higher paths are underutilized resulting in the uneven loads in the links available, on the cost of traffic resources. The development of MPLS addresses these problems with the use of constraint based routing (CBR).Signaling protocols are used to set up the paths for the packets to follow, these paths are commonly known as Label Switched Path (LSP) [5].
There are many protocols which can be used for the selection of paths but here we are concerned only on the signaling protocols that support Traffic Engineering, which are explained below:

- Constraint-based Routing Level Distribution Protocol (CR-LDP) it is the extension of the signaling protocol LDP. LDP is a control-driven LSP (known as hop by hop LSP or constraint-based LSP), the next hop here is determined either by looking up into the forwarding table of the LSR or control policy used. The control policy may be implemented by some application or the operators. CR-LDP is extended from LDP with the additional support to explicitly route the information about the traffic parameters for the reservation of the resources along the LSPs. CR-LDP and LDP are both hard state protocol as it sends the signaling messages only once without refreshing. It uses UDP (User Datagram Protocol ) for the peer discovery and TCP for rest of the process like session, advertisement and label request messages. DiffServ (Differentiated Services) as well as the operator configurable QoS (Quality of Service) are supported by CR-LDP [6].

- Resource Reservation Protocol (RSVP) uses the direct routes to set up the CR-LSPs. It uses UDP for resource reservation and label allocation. RSVP supports Integrated Service (IntServ) model of QoS. The Traffic Engineering extended version of RSVP known as RSVP-TE supports loop detection, periodization, reordering of path and strict and loose CR-LSPs. The path message is sent by the sender towards the destination to install the path state in each node. When the path message reaches the destination, the destination reply with the Resv message which reserves the resources as defined by the destination in the nodes and maintains the QoS parameters. Path and Resv message are refreshed periodically in RSVP which leads to the scalability problem in case of large traffic flow [5].

## 2. MPLS Operation

As shown in Fig.(1), Entire MPLS network can be divided into two parts namely MPLS edge and MPLS core. MPLS edge is the boundary of the MPLS network consisting of ingress and egress routers. MPLS core encompasses intermediate Label Switching Routers (LSRs), through which Label Switched Paths (LSPs) are formed. General terms associated with MPLS network and their meaning is specified as follow [7]:

- Label Edge Router (LER) – A router handles L3 lookups and is responsible for adding or removing the labels from the packets when they enter or leave the MPLS domain. Whenever a packet is entering or leaving MPLS domain it has to pass through LER router, when a packet enters into MPLS domain through LER which is called "Ingress router", or when a packet leaves the MPLS domain through LER which is called Egress router.

- Label Switch Router (LSR) – A router which is located in the MPLS domain and forwards the packets based on label switching is called LSR and usually this type is located in the provider cloud; as soon as LSR receives a packet it checks the look-up table and determines the next hop, then before forwarding the packet to next hop it removes the old label from the header and attaches new label.

- Label Distribution Protocol (LDP) – where the label mapping information is exchanged between LSRs. It is responsible in establishing and maintaining labels between switches and routers.

- Forward Equivalence Class (FEC) – set of packets where they have related characteristics which are forwarded with the same priority to the same path. This set of packets is has the same MPLS label. Each packet in MPLS network is assigned with FEC only once at the Ingress router.
- Label Switched path (LSP) – the path set by signaling protocols in MPLS domain. In MPLS domain there are number of LSPs that are originated at Ingress router and traverses one or more core LSRs and terminates at Egress router [7].

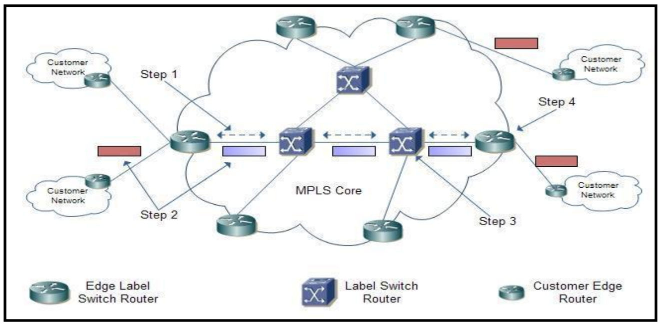


Fig. 1 MPLS Operation

## 3. VoIP performance metrics

VoIP performance is measured according ITU recommendations based on different parameters like (delay, jitter, and packet loss), these parameters can be changed and controlled within the acceptable range to improved VoIP QoS. Factors affecting QoS are briefly described in the following sections[8]:

- Jitter (Variation of Delay): In order for voice to be intelligible, voice packets must arrive at regular Intervals. Jitter describes the degree of fluctuation in packet access, which can be caused by too much traffic on the line. Voice packets can tolerate only about 75 Milliseconds (0.075 sec) but is preferred be 40 Milliseconds (0.040 sec) of jitter delay Equation (1) shows the calculation of jitter ($J$). Both average delay and jitter are measured in seconds .Obviously, if all ($di$) delay values are equal, then $D = di$ and $J = 0$ (i.e., there is no jitter) [9].

$$J = \sqrt{\frac{1}{N-1}\sum_{i=1}^{N}(di - D)^2} \qquad (1)$$

- Latency: As a delay sensitive application, voice cannot tolerate too much delay. Latency is the average time it takes for a packet to travel from its source to its destination. A person whose speaking into the phone called the source and the destination is the listener at the other end. This is one-way latency. Ideally, must keeping on the delay as low as possible but if there is too much traffic on the line or if a voice packet gets stuck behind a bunch of data packets (such as an email attachment), the voice packet will be delayed to the point that the quality of the call is compromised. The Maximum amount of latency that a voice call can tolerate one way is 150 Milliseconds (0.15 sec) but is preferred be 100 Milliseconds (0.10 sec). Equation (2) shows the calculation of delay where Average delay ($D$) is expressed as the sum of all delays ($di$), divided by the total number of all measurements ($N$) [9].

$$D = \sum_{i=1}^{N} di / N \qquad (2)$$

- Packet loss: is the term used to describe the packets that do not arrive at the intended destination that happened when a device (router, switch, and link) is overloaded and cannot accept any incoming data at a given moment. Packets will be dropped during periods of network congestion Equation (3) shows the calculation of packet loss ratio defined as a ratio of the number of lost packets to the total number of transmitted packets where $N$ equals the total number of packets transmitted during a specific time period, and $NL$ equals the number of packets lost during the same time period:

$$\textit{Loss packets ratio} = (NL/N)\times 100\% \qquad (3)$$

- MOS (Mean Opinion Score): MOS gives a numerical indication of the perceived quality of media received after being transmitted and eventually compressed using codecs. MOS is expressed in one number, from 1 to 5, 1 being the worst and 5 the best[10].

- End to end Delay**:** is the total transit time for packets in a data stream to arrive at the endpoint and it is inevitable in communication system. Delay time is one of the most important factors in determining the quality of a call [11].

## 4. Experimental Results

### 4.1 Proposed Scenarios

As shown in Fig.(2) and Fig.(3) respectively, the proposed model was implemented in two scenarios, one for sending voice using SIP-based IP (Scenario1) and the other model using LDP-based MPLS (Scenario2).
Scenario1: By using Traditional IP Network which use traditional routers and switches. The VoIP traffic (voice) is

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sent from source (client1) to destination (client3) as shown Fig.(2).
Scenario2: MPLS Network as shown in Fig.(3). The VoIP traffic (voice) is sent from source (client1) to destination (client3) but with using infrastructure devices which support MPLS.The main goal is to compare the performance of Voice traffic in the both networks by using performance metrics [12].

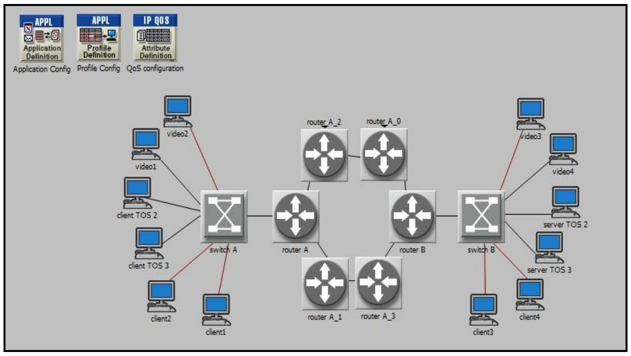


Fig.2 The Voice over IP Network Model

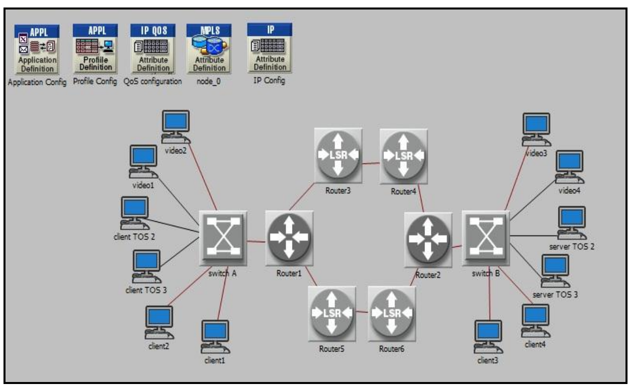


Fig.3The MPLS Network Model

Figure 3 shows the MPLS based scenario which consists of the following network elements:

- Two LERs (Router -1 and Router -2)
- Four LSRs (Router -3,Router -4,Router -5 and Router -6).
- Two VoIP stations (client -1 and client -3)
- Two switches (switch -A and switch -B)

## 4.2 Results and Discussions

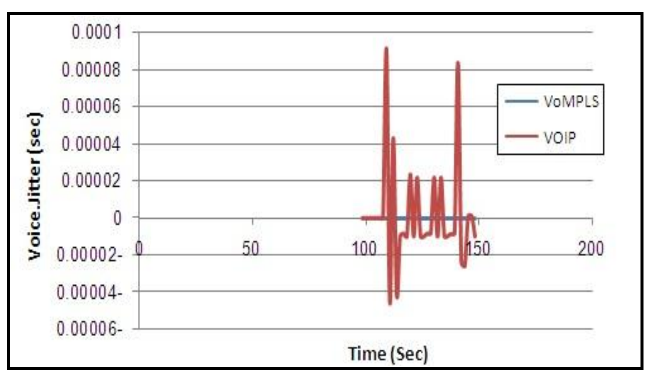
The performance of the two presented scenarios were compared based on the performance matrices, (parameters) such as: Voice delay (sec), Voice jitter (sec), Voice traffic sends (bytes/sec), Voice traffic received (bytes/sec) and MOS .The duration of simulation is 8 minutes and the results are obtained as shown in Fig.(4).Fig.(4-a), shows MPLS provides low values of jitter on comparison with VoIP EG at sec 109 the (Jitter = 0) using MPLS; whereas jitter using IP =0.000091 sec. It is noticed that voice jitter starts to rise at 0 sec in IP network up to 0.000091sec.It is observed from the Fig.(4-a) that there is an increase in the jitter for IP. At this time instant there is a voice packet drop .This variation at these time instants can be seen for the remaining performance metric.

on the other hand Fig.(4-b), which shows the end to end delay .it shows a great improvement& stability on the end to end delay of MPLS systems with respect of VoIP E.G. IP acceptable value up to (0.076833 sec) but using MPLS did exceed the time constraint (0.060026 sec). As explained in the above section, the end-to-end delay in a network is not advised to increase above the threshold value of 0.08sec. So that established VoIP calls are of acceptable quality.

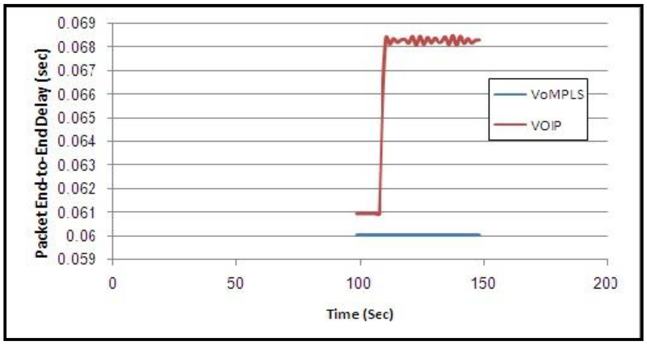
Fig.(4-c), regarding the voice traffic received it was shown that MPLS are more stable than VoIP E.G.at time 109s the voice traffic received using IP =15946.67(Bytes/sec) then varying to 16000(Bytes/sec), the voice traffic received using MPLS = 16000 (Bytes/sec).Voice packets start to drop from 109 second in the IP network, whereas no packets drop in MPLS network.

In simulation, the packet drop in IP network indicates that it cannot establish the VoIP calls with acceptable quality after 109 seconds.

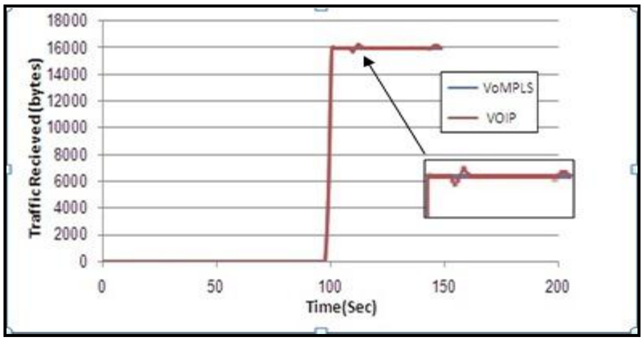
The VoIP calls established after 109 seconds exhibit packet loss. This cause loss of information and results in voice breaks and voice skips. Fig.(4-d), shows the voice traffic sent in the two cases =16000Bytes/sec and there is no great improvement. Finally Fig.(4-e), shows MOS value of the 2 Scenarios .it was found that the MOS value at 150sec the MPLS provides MOS value up to 4.36 while VoIP provides about 3.69 which improve a great improvement in comparison with VoIP.



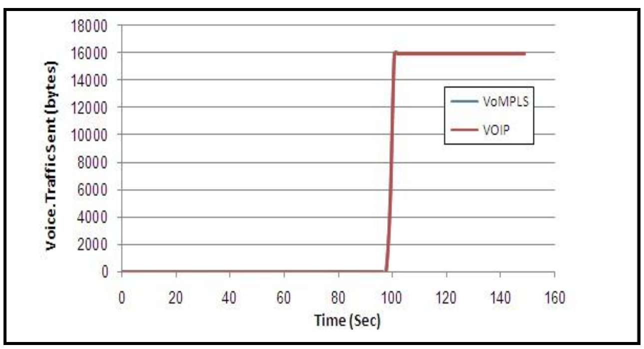

(a) Voice Jitter



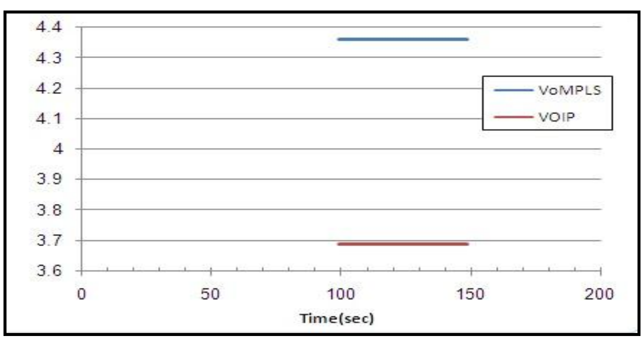

(b) Voice Packet end to end delay




(c) Voice Traffic Received (bytes/sec)




(d) Voice Traffic Send (bytes/sec)




(e) MOS

Fig.4 Comparison between the two network models

## 5. Conclusions

Next generation networks with multiple technologies offer different services to the user . In this study we make an extensive network simulations using MPLS compared with network using VoIP .We have analyzed several important parameters such as end-to-end delay, packet delay variation ,MOS, traffic send and traffic received. The experimental results using MPLS providing a great improvement in overall performance for voice traffic transmission and receive with lower voice packet delay, higher MOS value ,higher signal quality and lower voice jitter; therefore it can be concluded that implementing internet networks with support all customers for using MPLS is a very important field which need more and more research .

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